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LITERATURE SURVEY OF ACTIVE NOISE CONTROL SYSTEMS

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ABSTARCT: -- This paper will discuss various method of noise reduction for wireless communication network. Noise is an, unwanted and inevitable interference, in any form of communication. It is non-informative and plays the role of sucking the intelligence of the original signal. Any kind of processing of the signal contributes to the noise addition. A signal traveling through the channel also gathers lots of noise**.** It degrades the quality of the information signal**.** The effect of noise could be reduced only at the cost of the bandwidth of the channel, which is again undesired, as bandwidth is a precious resource. Hence to regenerate original signal, it is tried to reduce the power of the noise signal, or in the other way, raise the power level of the informative signal, at the receiver end this leads to improvement in the signal to noise ratio (SNR).

INTEX-TERMS: Active Noise Control, Filtered-X LMS algorithm, Filtered-U recursive LMS algorithm, Output whitening method, Matlab6.5.

1. INTRODUCTION

Acoustic Noise Control traditionally involves passive methods such as enclosures, barriers and silencers to attenuate noise. These techniques use either the concept of impendence change or the energy loss due to sound absorbing materials. These methods are however not effective for low frequency noise. A technique to overcome this problem is Active Noise Cancellation (ANC), which is sound field modification by elect acoustic means. ANC is an electro acoustic system that cancels the primary unwanted noise by introducing a canceling "antinoise" of equal amplitude but opposite phase, thus resulting in an attenuated residual noise signal as shown in Figure 1.

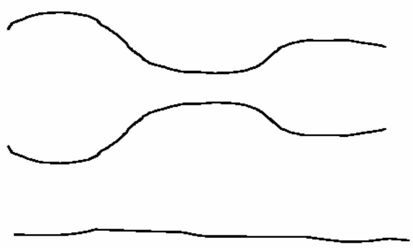

Fig. 1. Physical concept of Active Noise Control.

The design of ANC systems was first conceived in the 1930's by Lueg [1]. In the ensuing years, ANC has been the focus of a lot of research. An overview can be found in the tutorial paper by Kuo and Morgan [2] and also in the book by the same authors [3]. ANC systems are based either on *feed forward* control where a coherent reference noise input is sensed or *feedback* control [5] where the controller does not have the benefit of a reference signal. Further, ANC systems are classified based on the type of noise they attempt to cancel as either broadband or narrowband. A brief overview of the various approaches to ANC follows next.

2 BROADBAND FEED FORWARD ACTIVE NOISE CONTROL

These are systems that have a single secondary source, a single reference sensor and a single error sensor. The single channel duct acoustic ANC system shown in Figure.2 is an example of such a system

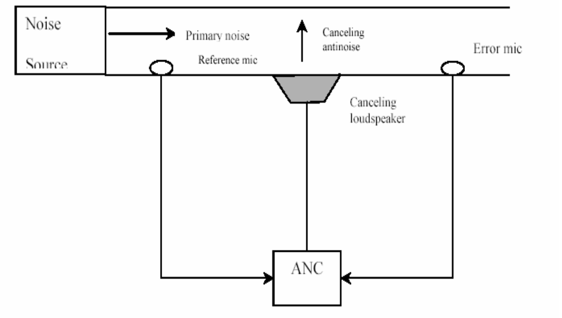

Fig. 2 Single channel broadband feed forward Active Noise Control.

This basic broadband ANC system can be described as an adaptive system identification framework as shown in Figure 3. Essentially, an adaptive filter W(z) is used to estimate an unknown plant P(z) which consists of the acoustic response from the reference sensor to the error sensor. The objective of the adaptive filter W (z) is to minimize the residual error signal e (n). However, the main difference from the traditional system identification scheme is the use of an acoustic summing junction instead of the subtraction of electrical signals. Therefore it is necessary to compensate for the secondary path transfer function S(z) from the output of the adaptive filter till the point where the error signal gets recorded.

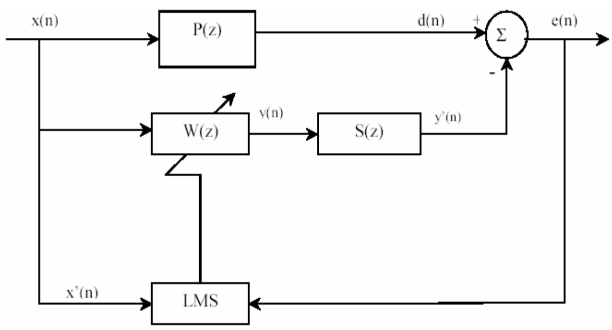

Fig.3. System identification view of ANC.

From Figure.3, we see that the z transform of the error signal is given by

$$E(z) = X(z)\left[P(z) - S(z)W(z)\right]$$

---------- (1)

Assuming that after convergence of the adaptive filter, the error signal is zero, W(z) is required to realize the optimal transfer function.

$$W(z) = \frac{P(z)}{S(z)}$$

----------------- (2)

The introduction of the secondary path transfer function in a system using the standard LMS algorithm leads to instability. This is because, it is impossible to compensate for the inherent delay due to S(z) if the primary path P(z) does not contain a delay of equal length. Also, a very large FIR filter would be required to effectively model 1/S(z). This can be solved by placing an identical filter in the reference signal path to the weight update of the LMS equation. This is known as the filtered-X LMS algorithm [5] [6]. The block diagram of an ANC system using the FXLMS algorithm is shown in Figure 2.4.

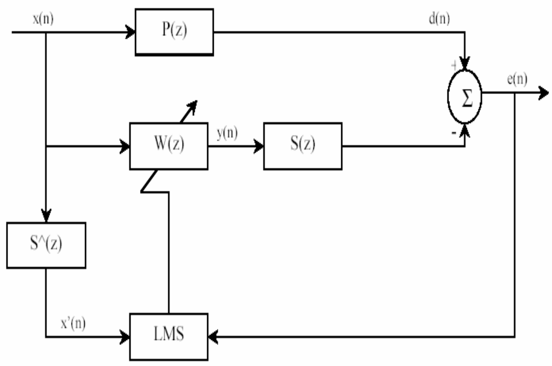

Fig.4. ANC system using the FXLMS algorithm.

A rudimentary explanation of the FXLMS algorithm is presented below. In figure 4, the residual error signal can be expressed as

$$E(n) = d(n) - s(n)*[w^T(n)x(n)]$$

--------- (3)

where s(n) is the impulse response of the secondary path S(z) at time n. Assuming a mean square cost function .(n) = E[e2(n)], the adaptive filter minimizes the instantaneous squared error)

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \frac{\mu}{2}\nabla\hat{\xi}(n)$$

Since ------------- (4)

$$\nabla\hat{\xi}(n) = -2\mathbf{x}'(n)e(n)$$

--------------- (5)

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\mathbf{x}'(n)e(n)$$

------------- (6)

In practical applications, the secondary path transfer function S(z) is unknown and must be estimated by an additional filter) (ˆ z S . Therefore, **x'**(n) =) (ˆ n S * **x**(n), where) (ˆ n S is the impulse response of) (ˆ z S . As shown by Morgan [7], the FXLMS algorithm seems to be remarkably tolerant to errors in the estimation of S(z) by the filter) (ˆ z S and within the limit of slow adaptation, the algorithm will converge with nearly 90?of phase error between) (ˆ z S and S(z) [7]. Therefore, offline modeling techniques can be used to model S(z) [3]. Nelson and Elliot [8] showed that the maximum step size that can be used with the FXLMS algorithm is given by

$$\mu_{\max} = \frac{1}{P_{x'}(L+\Delta)} \qquad \text{-------- (7)}$$

Where Px' = E[x'2(n)] is the power of the filtered reference signal and the number of samples corresponding to the overall delay in the secondary path. However, errors in estimating the secondary path transfer function will alter the stability bounds on [9]. A detailed analysis of the stability criterion is available in the literature[10] In the feed forward ANC system shown in Figure.3, the antinoise output of the speaker also radiates upstream to the reference microphone resulting in acoustic feedback and hence a corrupted reference signal x(n). Instability will occur if the open loop phase lag reaches 180 degree and the gain is greater than unity. This can be solved by using a separate offline adaptive feedback cancellation filter within the ANC system. Feedback can also be solved by using an adaptive IIR filter in place of the FIR filter in the ANC system. However, IIR filters are not unconditionally stable, as adaptation may converge to a local minimum and can have relatively slow convergence rates. A detailed analysis of adaptive IIR filters is available in the literature [11].

3. NARROWBAND FEED FORWARD ANC

Many noise sources are periodic in nature such as engines, compressors, motors, fans, etc. In such cases, direct observation of the mechanical motion using an appropriate sensor is used to provide an electrical reference signal which consists of the primary frequency and all the harmonics of the generated noise. The basic block diagram is as shown in Figure 5

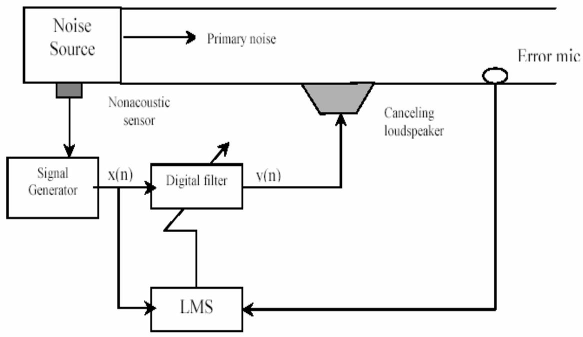

Fig.5. Narrowband feed forward ANC system.

This technique avoids the undesired acoustic feedback to the reference sensor, as well as nonlinearities and aging problems with acoustic microphones. The periodicity of the noise removes the causality constraint, as each harmonic can be controlled independently and a much shorter FIR filter can be used to model the secondary path. There are two techniques for narrowband ANC i.e. the waveform synthesis method, which uses an impulse train with a period equal to the inverse of the fundamental frequency of the disturbance. The second technique uses an adaptive notch filter with a sinusoidal reference signal.

4. FEEDBACK ACTIVE NOISE CONTROL

Feed forward ANC systems (broadband and narrowband) use a reference sensor to measure the primary noise signal, a feed forward adaptive filter and an error sensor to measure the residual error signal. However, in some applications, it is not feasible to have a sensor to measure or internally generate the error signal. This section describes a class of algorithms known as feedback ANC in which the reference signal is generated from the output of the error sensor. This is used in applications that combat spatially incoherent noise generated from turbulence, noise generated from many sources and propagation path induced resonance where no coherent reference signal is available.

5. NONADAPTIVE FEEDBACK ACTIVE NOISE CONTROL

Olson and May [4] first introduced a nonadaptive feedback ANC system that used an amplifier carefully matched to the response of the error sensor and the secondary source. Hong and Eghtesadi [12] studied the application of feedback ANC for noise reduction in a duct. Veit [13] , Carme [14] and Wheeler [15] studied the application of feedback ANC for noise compensation in personal hearing protectors. The basic block diagram of a classical feedback ANC system is as shown in Figure 6

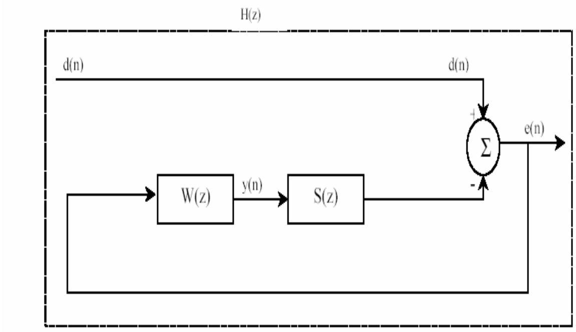

Fig.6. Classical feedback ANC system.

In Figure.6 above, d(n) is the primary noise at the error sensor location, e(n) is the residual noise, y(n) is the secondary antinoise signal, W(z) is the transfer function of the controller and S(z) is the transfer function of the secondary path. Under steady state conditions, the z-transform of the error signal can be expressed as therefore the closed loop transfer function H(z) from the primary noise to the error signal can be expressed as

$$H(z) = \frac{E(z)}{D(z)} = \frac{1}{1 + S(z)W(z)}$$

----------------(8)

From equation 8 the power spectrum of the error signal is given by

$$S_{ee}(w) = \frac{1}{\left|1 + S(w)W(w)\right|^2} S_{dd}(w)$$

----------------(9)

where See(w) and Sdd(w) are the power spectra of the error signal e(n) and the reference noise d(n) respectively. Therefore, in order to minimize See(w), we need to minimize 2) ( ) (1 *z W z S* + , or the gain of S(z)W(z) should approach infinity. If the frequency response of S(w) is flat, then the gain of W(w) can be increased without limit so that the overall transfer function of the feedback loop becomes marginal. However, this is rarely the case as the response of the secondary source introduces a significant phase shift and there is some propagation delay from the output of the control filter to the error sensor. These effects introduce a phase shift in S (w) that increases with frequency. As the phase shift approaches 180°, the desired negative feedback becomes positive feedback leading to instability. Therefore as the frequency and phase shift increase, the gain of W (w) should decrease. Hence it is possible to design an inverting amplifier W (w) provided the gain is not large enough to make the net loop gain greater than unity when the phase shift is 180°. Therefore, if

$$S(w)W(w) = G(w)e^{j\phi(w)}$$

--------------- (10)

$$\left|1 + S(w)W(w)\right|^2 = 1 + G^2(w) + 2G(w)\cos\phi(w)$$

---------- (11)

Given a secondary path S(w), W(w) needs to be chosen such that the net gain G(w) is Maximized when -180°< f(w) < 180°. A more detailed explanation of the design of feedback ANC system is available in the literature [3][13][14].

6. SINGLE CHANNEL ADAPTIVE FEEDBACK ACTIVE NOISE CANCELLATION

The adaptive single channel Active Noise Cancellation system was first proposed by Eriksson[16] and then extended to the multi channel scenario by Popovich [17][18]. This technique is generally viewed as an adaptive feedforward ANC system that in effect synthesizes its own reference signal. Under certain conditions, the system can also be interpreted as an adaptive predictor [19].

In the feedback ANC system shown in Figure.6, the primary noise signal d(n) is not available. Therefore, the main idea of an adaptive feedback ANC system is to regenerate the reference signal d(n) from the error signal. From Fig 6, we can see that the primary noise can be expressed in the z-domain as

$$D(z) = E(z) + S(z)Y(z)$$

--------------- (12)

Where E (z) is the residual error signal obtained from the error signal and Y (z) is the output of the adaptive filter. The secondary path transfer function S(z) can also be estimated as) (ˆ *z S* . Thus we can estimate the primary noise d(n) and use this as a synthesized reference signal x(n) as follows

$$X(z) \equiv \hat{D}(z) = E(z) + \hat{S}(z)Y(z)$$

----------- (13)

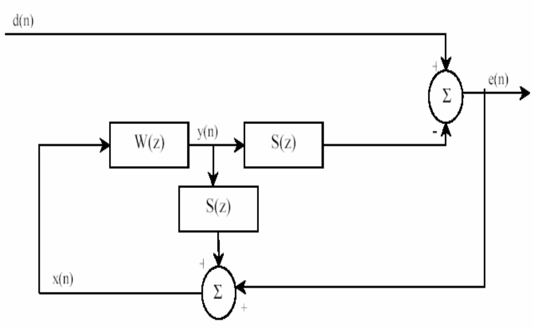

***Fig.7. Adaptive feedback ANC system using synthesized reference signal**.*

A complete block diagram of the broadband ANC system is shown in Figure 8

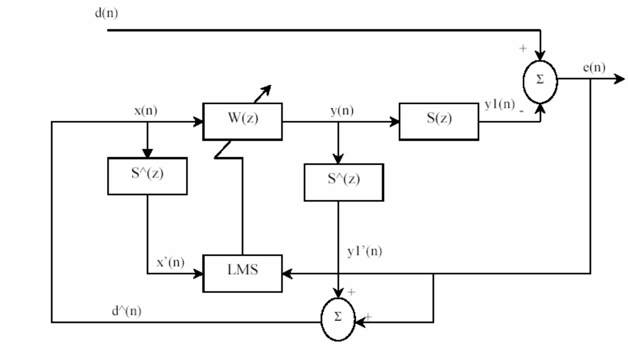

Fig 8. Broadband feedback Active Noise Cancellation using the FXLMS algorithm.

From Figure 8, we can see that the reference signal x(n) which is synthesized from the error signal can be expressed as

$$x(n) \equiv \hat{d}(n) = e(n) + \sum_{m=0}^{M-1} \hat{s}_m y(n-m)$$

--------(14)

Where *m s*ˆ , m = 0, 1, ……….M-1 is the Mth order FIR filter ( ) (ˆ *z S*) used to approximate the secondary path transfer function. This estimation can be performed either online or offline. The secondary signal y(n) is generated as

$$y(n) = \sum_{l=0}^{L-1} w_l(n) x(n-l)$$

-------(15)

Where w(n), l = 0, 1, ………L-1 are the coefficients of the Lth order adaptive FIR filter W(z) at time n. These coefficients are updated by the FXLMS algorithm as

$$w_l(n+1) = w_l(n) + \mu x'(n-l)e(l)$$

--------------- (16)

where is the step size and the filtered reference signal x'(n) is given by From equations 14 and 15, we can see that x(n) = d(n) if) (ˆ *z S* = S(z). Assuming that this condition is satisfied, and then the adaptive feedback ANC system in Fig.8 can be transformed into the feed forward ANC system in Figure 4. The adaptive filter W(z) can be commuted with the secondary path transfer function S(z) if the LMS algorithm has slow convergence, i.e. the step size is small [5]. Further, if we assume that the secondary path S(z) can be modeled as a pure delay i.e. S(z) = z-., then the feedback ANC system is equivalent to the standard adaptive predictor as shown in Figure.9.

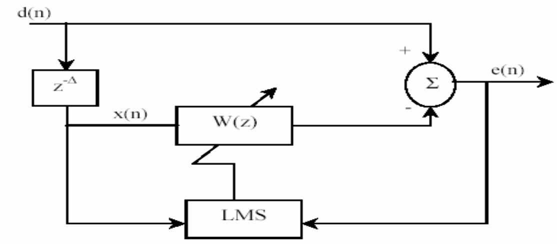

Fig .9. Feedback Active Noise Cancellation algorithm as an adaptive predictor.

Thus the feedback ANC algorithm acts as an adaptive predictor of the primary noise d(n) to minimize the residual error noise e(n). Hence the performance of the algorithm depends on the predictability of the primary noise d(n). A detailed analysis of the feedback ANC algorithm as an adaptive predictor is available in the literature [19]. We know that the error signal in Figure 8 can be expressed in the z-domain as

$$E(z) = D(z) - S(z)Y(z)$$

------------ (17)

Where

$$Y(z) = W(x)[E(z) + \hat{S}(z)Y(z)]$$

---------- (18)

Rearranging equation 18 we get

$$[1-\hat{S}(z)W(z)]Y(z) = W(z)E(z)$$

---- (19)

$$Y(z) = \frac{W(z)E(z)}{1-\hat{S}(z)W(z)}$$

---- -- (20)

Substituting equation 19 in 20, we get

$$E(z) = D(z) - S(z)\left[\frac{W(z)E(z)}{1-\hat{S}(z)W(z)}\right]$$

----- ---- (21)

$$\left[1+\frac{S(z)W(z)}{1-\hat{S}(z)W(z)}\right]E(z) = D(z)$$

------------- (22)

$$E(z) = \frac{D(z)}{\left[1+\frac{S(z)W(z)}{1-\hat{S}(z)W(z)}\right]}$$

------- (23)

$$E(z) = \frac{D(z)\left[1-\hat{S}(z)W(z)\right]}{1+[S(z)-\hat{S}(z)]W(z)}$$

------------ (24)

$$E(z) = D(z) - S(z)W(z)D(z)$$

--------- (25)

Therefore, the overall transfer function of the feedback ANC system from d(n) to e(n) is given by Therefore under ideal conditions, the feedback ANC system is transformed to a feed forward ANC system. For certain applications where the noise to be cancelled is narrowband, the waveform synthesis method can be used. In this method, the regenerated reference signal x(n) is used to synthesize a low frequency component that has a repetition rate locked to the fundamental driving frequency of the primary noise source. The reference signal estimate is fed directly to a phased lock loop, which generates a synchronization pulse for a waveform synthesizer. A more detailed explanation is available in the literature [20].

A number of alternate schemes have been proposed for feedback ANC. Oppenheim and Zangi [21] proposed a feedback ANC scheme based on the block Expectation Maximize algorithm. Openheim et al [22] proposed a scheme based on the RLS algorithm. However, these algorithms have been generated with ideal conditions and initial conditions need to be very carefully generated to ensure that the algorithm converges. Eriksson et al [23] proposed a generalized recursive ANC scheme that uses three adaptive filters that accurately model the primary path, the feed forward path and the feedback path for the filtered X or filtered U algorithms. Performance analysis of all these algorithms have shown that the noise attenuation is very noticeable in the lower frequency regions below 1kHz and deteriorates very rapidly as the center frequency of the noise increases [3][19][24].

7. HYBRID ACTIVE NOISE CONTROL SYSTEMS

A combination of the feed forward and feedback ANC schemes is known as the Hybrid ANC scheme. Here, the canceling signal is generated based on the inputs of both the reference sensor and the error sensor. This method was first proposed by Swanson [27]. The motivation behind this method is to increase the correlation between the primary noise and the signal picked up by the reference sensor. Since the error sensor is generally placed downstream from the source of the primary noise and the reference sensor is places as close as possible to the primary source. However, it is often the case that the reference sensor may not pick up all the acoustic cues of the primary noise source and this can be rectified by using the signal at the error sensor also to generate the canceling output [19]. The block diagram of the Hybrid ANC scheme is as shown in Figure 10

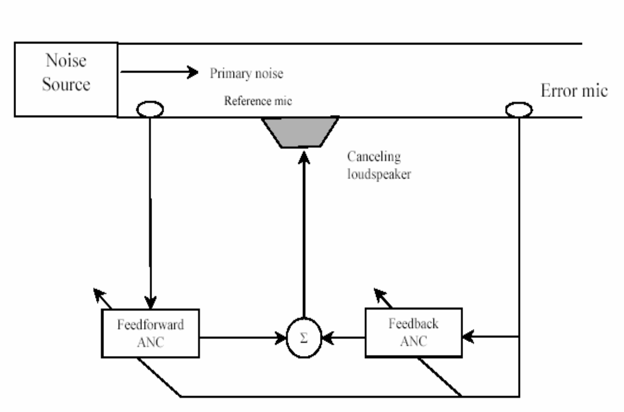

Fig 10. Active Noise Cancellation system with both feedback and feed forward Control loops

The system in Figure 10 can be configured with the feed forward path as any one of the following:

1. Filtered-X LMS algorithm
2. Filtered-X LMS algorithm with feedback cancellation
3. Filtered-U recursive LMS algorithm

The feedback path can be configured to be

1. Classical feedback ANC
2. Feedback ANC using Filtered-X LMS algorithm
3. Output whitening method [28]

The canceling signal fed to the speaker is generally an unweighted sum of the outputs of both algorithms. Vijayan [19] showed that the hybrid scheme was better at canceling broadband noise, when compared to the feed forward or feedback schemes by themselves. Furthermore, she showed that it is possible to reduce the length of the filters in the hybrid scheme. Feedback Active Noise Cancellation has been used in many applications. It has been used in hearing protectors such as headsets [13][14][15][24] along with other passive methods to reduce noise especially on the factory floor and in aircraft cockpits, etc. There are many commercially available products that use nonadaptive ANC schemes for noise cancellation in headsets. ANC has also been combined with other applications such as headsets for communication purposes [24], in integrated hands-free kits for cellular phones [25] and has been implemented using the speaker and microphone available in the cellular phone itself [26]. In all these applications, the generated antinoise needs to be combined with the signal of interest such as the received speech signal or the audio signal from an external sound source, etc. Hence the error microphone will pick up the residual error source but also the signal from the external source. Therefore, it is possible that the antinoise signal generated by the ANC system can degrade the quality of the desired signal as well.

8. MULTI CHANNEL ACTIVE NOISE CANCELLATION

In many applications, it is desirable to cancel noise at several locations in a three dimensional space. Single channel systems are effective when the area of interest is restricted and there is only a single primary source that can be accurately located and a single "quiet zone" where the error sensor needs to be located. However, many practical applications involve relatively large multidimensional spaces where the noise source cannot be accurately pointed to be at one single location. The complexity of multi channel ANC in a multi dimensional space is however significantly higher and the system needs to be carefully ported to a practical real world application.

The best known applications of multichannel systems are in the control of exhaust noise in automobile cabins as well as in the cabins of heavy equipment such as earth movers, flight cabins, etc [29][30]. These algorithms generally tend to be single reference, multiple error sensors. The idea is to use a single nonacoustic sensor to generate the periodic reference signal and to minimize the sum of the squares of the outputs of a large number of equally spaced error sensors. Generally, multi channel algorithms are necessary when the area in which ANC needs to be performed becomes larger. Elliott[31] proposed a multi channel FXLMS algorithm to cancel the noise created by rotating machinery using multiple error sensors and a single nonacoustic reference sensor. Signal processing structures have also been proposed for multiple references, multiple outputs broadband feed forward ANC [32]. Multichannel ANC usually requires careful research into the acoustic characteristics of the environment in which it is being implemented. Usually, the secondary path transfer functions have a nonminimum phase, i.e. have zeros outside the unit circle due to the reverberances in the three dimensional space. The general strategy is to minimize the total energy in the multidimensional space i.e. the sum of the outputs of all the error sensors. Therefore, the location of these sensors is very important so that it represents the sum total of the energy in the multidimensional space. The multichannel feed forward ANC system can thus be viewed to be a combination of single channel feed forward systems, with the exception that there are multiple secondary paths from each of the adaptive filters to each of the error sensors.

Multichannel feed forward algorithms are also generally prone to feedback due to the larger number of error and reference sensors. It has also generally been found that IIR adaptive filters are more effective than FIR filters in multi channel systems [33]. Multi channel systems have also been implemented using the feedback ANC algorithm using either a K X 1 system with K reference sources and a single error source or a K X M system with K reference sensors and M error sensors [3].

9. FREQUENCY DOMAIN AND SUBBAND ACTIVE NOISE CANCELLATION

9.1. Frequency domain ANC*:*

The feed forward ANC system has also been implemented in the frequency domain. In this implementation, the reference signals x (n) is first stored in an L point buffer and then transformed into the frequency domain signal using an L point FFT. The FFT spectrum is then multiplied by the appropriate adaptive weights to generate a frequency domain output signal which is then retransformed into the time domain using an L point IFFT. This is as shown in Figure 11.

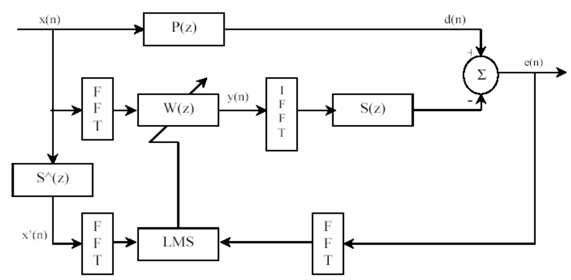

Fig .11. Frequency domain feedback FXLMS algorithm.

The system shown in Figure 11 was first proposed by Shen and Spanias [34] and subsequently extended to the multiple channel case [35]. Reichard and Swanson [36] implemented a frequency domain feed forward FXLMS structure with online system identification. Feintuch et al [37] showed that as long as the frequency band of interest was limited, it was not really necessary to estimate the transfer function of the secondary path before adaptation and it was only necessary to know the delay introduced by the transfer function in the band. The major drawback of the frequency domain algorithm is that it processes the data block by block instead of sample by sample. Thus there are L samples of delay between the input of the reference signal and the output of the secondary antinoise signal. This delay would be tolerable for very low frequency periodic noise. However, in the case of broadband noise, the delay is a major shortcoming. The delay caused when the frequency domain FXLMS algorithm is adapted to the feedback ANC system becomes 2L as there is a further L sample delay while the reference signal is regenerated. Hence the frequency domain implementation of the feedback FXLMS structure is highly inappropriate.

9.2Sub band Active Noise Cancellation*:*

There are a number of applications in which ANC has been used to cancel broadband noise. In order to effectively cancel broadband noise, it is essential that adaptive filters with hundreds of taps be used. However, these are not only computationally intensive, but also display very slow convergence. Moreover, ANC has also been used in conjunction with other applications. In this scenario where the speaker is used not only to play the antinoise signal but also a desired signal from a secondary source, the error sensor will pick up both the residual noise as well as the desired signal. Hence it is probable that the generated antinoise signal will degrade the quality of the desired signal as well. Sub band adaptive filtering techniques have previously been proposed to solve the problems for acoustic echo cancellation, signal enhancement, etc. [38]. Morgan and Thi [39] adapted the same idea for feed forward Active Noise Control. The proposed structure was very similar to the frequency domain structure proposed earlier, i.e. the adaptive weights are computed for each sub band (FFT bin) separately, and then transmitted to an equivalent wideband filter. However, it differs from the frequency domain structure in that the actual processing of the sub band signal takes place in the time domain.

This technique is computationally intensive as it is required to take a polyphase FFT of the error and filtered reference signals as well as an inverse FFT of the filter weights. It is still less intensive than an equivalent wideband adaptive filter [40]. Park et al [41] proposed a modification to the sub band adaptive filter architecture that decomposed the secondary path transfer function into a series of sub bands as well. Both these studies showed that the sub band technique had a significantly better convergence when compared with the traditional wideband FXLMS algorithm. The sub band technique is all the more important when ANC is used in conjunction with another application. Hessian and Campbell [42] studied the application of sub band ANC to speech that had been corrupted with automobile noise. Their technique used two separate channels, with the sub band technique applied separately on each channel. These sub band techniques however, do not translate well to the feedback FXLMS system due to the need to buffer the reference signal and the error signal. A number of techniques has been proposed for sub band adaptive filtering using filter banks. Vitterli and Gilloire [43] and Use itch and Orchard [44] separately proposed a sub band filtering scheme where the expected signal and the error signal were divided into sub bands using a filter bank and each sub band was then processed independently. The conditions for the filter bank were researched by Petraglia and Alves [45] . Alves et al [46] studied the convergence properties of the sub band adaptive filter structure. The filter bank method is more suitable for sub band feedback ANC as the processing can be done on a per sample basis. The next chapter presents the proposed sub band feedback Active Noise Cancellation system.

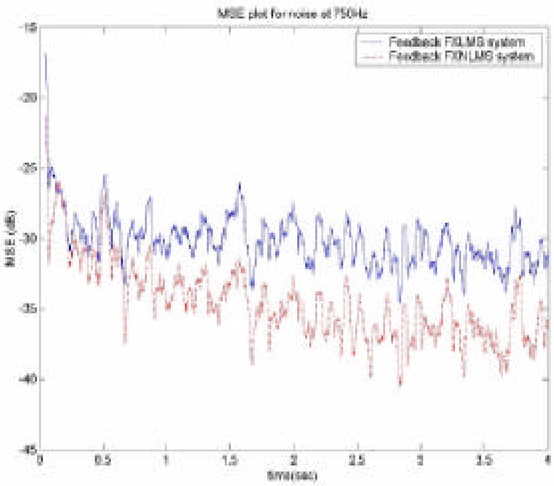

Fig 12. MSE plot for colored noise at 750Hz.

Frequency of primary noise signal	Noise attenuation of single band algorithm	Noise attenuation of subband algorithm
0-100Hz	14.4dB	16dB
500-750Hz and 1500-1750Hz	5.66dB	10.03dB
750-1000Hz and 2000-2250Hz	6.31dB	11.9dB
3000-3250Hz	9.7dB	15.79dB

Table 1 Noise attenuation of sub band system for noise at different frequencies.

10. CONCLUSION AND FUTURE WORK

This paper was discussed different method of noise reduction for wireless communication network. Finally the Sub band Active Noise Cancellation method was more accurate and efficient as compare to other method. This method was simulated by using Matlab software only. In future this method is implemented to DSP processor kit for real-time application purpose.

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